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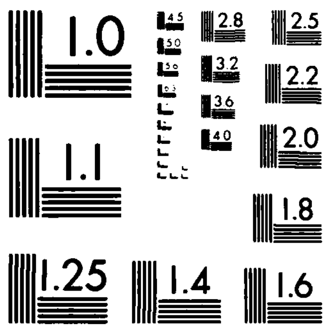
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AUDIO SIGNAL MANAGEMENT TECHNIQUES

HRB-Singer, Inc.

A. Penny Anderson, James K. Lane and Burl K. Pudliner

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ROME AIR DEVELOPMENT CENTER
Air Force Systems Command
Griffiss Air Force Base, NY 13441

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This report documents the design of an Exploratory Development Model (EDM) which will be used to study audio signal management techniques. The system was designed for the PDP 11/70 computer system and includes both software and hardware. The computer software design is documented in a software system specification and listings. The hardware is an Audio Distribution Network (ADN) which allows great flexibility to route audio signals. Documentation of the ADN is provided.

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Audio Signal Management Techniques (ASMT)
FINAL TECHNICAL REPORT

1.0 INTRODUCTION

1.1 GENERAL

This Final Technical Report (A003) is submitted in response to Rome Air Development Center (RADC) Contract No. F-30602-81-C-0300, Audio Signal Management Techniques.

1.2 PROGRAM OBJECTIVE

The objective of the Audio Signal Management technical program was to design and develop an Exploratory Development Model (EDM) Audio Signal Management System (ASMS). This system is to be used to test and evaluate present and future voice data entry algorithms, processing techniques, and hardware modules. The ASMS consists of internal functions implemented on the RADC PDP 11/70 computer, external functions implemented in stand-alone hardware devices, an Audio Distribution Network (ADN) for shaping and routing audio signals, and an ADP Data entry communication interface/keyboard translator with HP 2645A terminal for function control and transcription.

1.3 OUTLINE OF REPORT

This report will first review the three phases of design through which this project has passed. The first design was that described in the Interim Design Plan dated 7 April 1982; the second described in Section A of the Addendum to the Interim Design Plan, dated 10 May 1982; and the third, Section B of the same Addendum. The system demonstrated at RADC on 28 October 1982 will be described showing the relationship between the requirements specified in Section B of the Addendum and the delivered system. Finally, several comments will be provided for meeting the full complement of functions as described in the Addendum Section A.

2.0 REVIEW OF INTERIM DESIGN PLAN

An engineering investigation into possible solutions to the stated requirements of the ASMS revealed that 1) the PDP 11/70 was not adequate for the expected throughput requirements, and 2) an alternative design solution would be too expensive to complete within contract cost. These shortcomings were discussed during a Technical Review on 29 January 1982. As a result, HRB-Singer was directed to redesign the system to sustain the expected throughput requirements.

The resulting redesign was described in the Interim Design Plan, dated 7 April 1982. That design called for the use of microprocessors to offload from the PDP 11/70 many of the internal functions required. Briefly, one microprocessor per channel was to be included for the input internal functions of GSE, PSE, and ADPCM, and the output internal functions of ADPCDM, silence reconstruction, and speech rate change; GFE LPA's would still be used for input A/D and output D/A. Thus the proposed hardware and software configuration was comprised of an LPA for four-channel input A/D conversion; software for the four Intel 8086 microprocessors to perform the input internal functions; software for the PDP 11/70 to transfer converted values from the LPA to the microprocessors and preprocessed values from the microprocessors to disk; software for the microprocessors to perform the output internal functions; software for the PDP 11/70 to transfer stored values from the disk to the microprocessors, and transfer postprocessed values from the microprocessors to the D/A converter; a second LPA to perfrom the D/A conversion; software for communication between the 8085 based ADP Shorthand box and PDP 11/70; and an audio distribution network for routing of signals, engaging the external functions, and impedance matching.

Though the microprocessor-supported design was known to be complex, it offered the prospect of successfully meeting the throughput requirements stated in the SOW.

3.0 DESIGN REVIEW

A Design Review of the Interim Design was held on 15 April 1982. It was concluded that the SOW requirements had been overstated, and that the interim design was too complex and costly. Two means of simplifying the design were identified. First, the requirement for four simultaneous playback channels was replaced by the requirement for one playback channel, selectable from among the four input channels. Second, the throughput requirements were relaxed, so that the microprocessors could be eliminated from the design.

With these two modifications to the requirements, the ASMS was redesigned, with the resulting designs described in the Addendum to the Interim Design Plan, dated 10 May 1982. That Addendum included two sections. Section A described a SOW-compliant (fully-implemented) design, with all internal functions performed centrally in the PDP 11/70. Section B described a design which implemented a subset of the functions described in Section A, and which could be implemented within the time and funds constraints of the contract.

A more detailed review of the Section A and Section B designs follows.

4.0 REVIEW OF ADDENDUM SECTION A DESIGN

Section A of the Addendum to the Interim Design Plan for the ASMS describes a SOW-compliant "test bed" system. The design called for the centralization to the PDP 11/70 of each of the input and output internal functions, while maintaining hardware and external functions similar to those of the Interim Design.

The hardware configuration of this design differed from the original design in two ways. First, the parallel 8086 microprocessors were eliminated, since their functions were transferred to the PDP 11/70. Second, only one channel of D/A was provided to the Audio Distribution Network (ADN). Since it is not possible to transcribe more than one channel simultaneously, the selection of which one of the four input channels to playback is under operator control.

The external functions provided as GFE, and their interface with the ASMS, were the same as the original design. That is, the Speech Enhancement Unit (SEU) could be connected across any analog signal, and its software downloaded on command from the operator terminal; the Keyword Recognition Unit (KRU) could similarly bridge any analog signal, and be controlled from the operator terminal; projected external functions (e.g. Speaker Identification, Language Identification and Voice Training) were similarly provided ports on the ADN and controlled from the operator terminal.

The internal functions provided in this design were centralized to the GFE PDP 11/70; all software functions required by the SOW were included. Briefly, the functions provided, in the expected sequence of processing, were: control of A/D conversion via the LPA Fortran Support routines; Gross Silence Editing (GSE); Precise Silence Editing (PSE); Data Compression to 4 bits by Adaptive Pulse Code Modulation (ADPCM); Data Storage on disk; Data Retrieval from the disk; Data Decompression to 12 bits by ADPCM decoding; Audio Playback start and stop; PSE reconstruction; selective recall or audio looping or instance recall; Speech Rate Change; and D/A control via the Fortran LPA support routines. In addition, code on the PDP 11/70 would be created, and the code in the ADP Shorthand Unit modified, to control and display the activity of each of the internal functions.

The centralization of these internal functions had one major predictable consequence: throughput failures. The more thorough analysis of throughput considerations presented in Section A of the Addendum suggested that, if all internal functions were implemented, under even the best of conditions (minimum algorithms, 50% silence) only one channel could be recorded and played back simultaneously. If either the implemented algorithms were longer, or there was less silence on the channel, data would be lost. Attempts to simultaneously record additional channels would, of course, present additional throughput stress.

One method of evading the predicted throughput failure of the fully-implemented system was to implement a subset of those functions. A description of one such subset follows.

5.0 REVIEW OF ADDENDUM SECTION B DESIGN

Section B of the Addendum to the Interim Design Plan describes an implementation of the ASMS which provides a subset of the internal functions described in Section A. By implementing only a subset of those capabilities, it was hoped that a system could be provided which had no throughput failure for simultaneous recording and playback on at least one channel, and which provided a software architecture compatible with subsequent implementation of one or more of the additional functions described in Section A.

The hardware configuration of this design is the same as that for Section A. At the heart of the hardware configuration is the ADN. It provides input ports for four input audio channels and one D/A channel; four output ports to A/D converters; input and output ports for engaging the external functions; and a mixer for comparing audio input with D/A output. VU meters are provided to monitor input channel activity; and amplification and padding provided where necessary to match impedances between input and output ports.

The external functions are provided ports on the ADN as in the previous design. However, the capability to control these external functions from the ASMS operator console has been deleted. Thus, the KRU needs to be operated from its own terminal, and the SEU can only be used if internal downloading is provided.

The internal functions to be provided by the software system for the PDP 11/70 are as follows: control of A/D conversion via the LPA Fortran support routines; truncation of the 12-bit converted samples to 6 to 10 bits, as selected; storage of data on the RM03 disk; retrieval of the stored data from the disk; audio playback start and stop; selective recall of portions of the stored data; and control of D/A conversion via the Fortran LPA support routines. Control of these internal functions, and the updated display of the activity of the system, is provided in new code for both the PDP 11/70 and the 8085 based ADP Shorthand Unit.

Many of the internal functions of the previous design are deleted in the current design; by reducing the number of transforms applied to each sample, the throughput requirements for one channel of simultaneous recording and playback were expected to be attained. The functions not implemented in the current design are: GSE; PSE and silence reconstruction; ADPCM and its inverse decompression; audio looping and instance recall; and speech rate change. Though none of these are implemented, the software design includes consideration of their eventual incorporation into the software architecture.

6.0 DESCRIPTION OF IMPLEMENTED ASMS

After the design described in Section B of the Interim Design Plan was approved, hardware and software design, development, integration, and testing proceeded. The system was demonstrated and accepted by 28 October 1982.

The Hardware Technical Report (Appendix A) gives a detailed description of the demonstrated hardware configuration. The Software Technical Report (Appendix B) describes the overall software architecture, the individual software tasks, interactions among tasks (shared data files, global common event flags, global common areas), development methods, system integration and extensibility of the software, and system operating procedures. What follows is a brief description of capabilities of the ASMS installed at RADC.

The ASMS enables an operator to record from any one of four audio channels, and simultaneously playback that or any other channel. The operator's console displays which channel is being actively recorded, and the starting times of recorded data files, if they exist; it also displays which channel is being played back, the starting time of that channel, and, if an active recording channel is being played back, the number of seconds the playback is behind. The operator may start and stop recording and playback on any channel at any time, and can choose to play back any desired portion of any active or existent audio file. A message field on the console informs the operator of the current ASMS status. The operator may use the audio routing patch panel and mixer volume controls to selectively listen to any pair of audio signals, enable or disable AGC on any input channel, and engage any external function. In addition to co trolling the process of recording and

playback, the operator can use the ADP feature of the console to transcribe the audio signals, and the ADP command subset to control the transcription process.

Thus, the implemented ASMS enables an operator to use the console commands and adjoining Audio Distribution Network to selectively record and playback from any of four channels, selectively listen to any pair of audio signals, and transcribe and store the selected data.

7.0 SECTION B REQUIREMENT MET BY IMPLEMENTED ASMS

Since only one LPA was available for system integration and test, the peripheral hardware restricted the system to simultaneous recording and playback of only one channel. Within this limitation, the ASMS at RADC met all requirements described in Section B of the Addendum to the Interim Design Plan. The installed ASMS listing is provided in Addendum C.

8.0 EXTENDING ASMS TO ORIGINAL SOW REQUIREMENTS

The ASMS demonstrated at RADC was designed to provide a software architecture for future expansion of the full capabilities described in Section A of the Addendum to the Interim Design Plan. This section will discuss the probabilities of success for implementing 1) operator control of external functions; 2) Audio Looping and Instance Recall; 3) Speech Rate Change; 4) GSE, PSE, & ADPCM; and 5) expansion from one to four channels of simultaneous recording.

8.1 OPERATOR CONTROL OF EXTERNAL FUNCTIONS

This includes the capability for the operator to download the SEU, and control the KRU from the ASMS console. Unless additional information is provided to indicate otherwise it can be assumed that the addition of these features should entail little system overhead.

8.2 AUDIO LOOPING AND INSTANCE RECALL

These functions need to be applied to only the single output channel, and their functions are quite similar to that of starting playback at an arbitrary point in time, which already provided in the current implementation. It appears that these functions could easily be implemented within the current software architecture.

8.3 SPEECH RATE CHANGE

This function, similarly, needs only be applied to the single output channel. Since the Playback task currently has approximately 4K words of free task address space, it appears that the algorithm could be implemented, depending on its complexity. For playback speeds greater than one, of course, there will be the additional input QIO overhead for reading up to twice as fast the input data file.

8.4 ADPCM, GSE, & PSE

These three input functions will be described together since the implementation of each is anticipated to have similar system throughput impacts.

Each function will lead to more efficient storage of data, at the cost of increased processing on each input sample; i.e., the functions will tend to shift system performance from being I/O-bound to compute-bound. The increased efficiency of storage of the compressed samples may be used to decrease the size of I/O buffers in order to release additional program space within the input tasks' task address space. Since the input tasks currently have little free task address space, the latter alternative will probably be necessary.

The first of these functions to be implemented should be the ADPCM. It will have a predictable impact on storage packing density (2 or 4-fold increased density), and will to some extent simplify the Record and Playback tasks, since samples will be uniformly 4 bits, rather than variably 6 to 10 bits. The packing density achieved with ADPCM will represent the worst-case storage density when both GSE and PSE algorithms fail.

The GSE and PSE algorithms should be the last ones attempted, since their effects on storage and system throughput are the least predictable. For any given speaker, the PSE function will tend to have more uniform effects than GSE; over a range of speakers, however, its effects will probably be as unpredictable as those for GSE.

The three functions of ADPCM and its inverse, PSE and its inverse, and GSE offer the potential for extremely high storage density, at the cost of sharply increasing compute time, and increasing strain on the already tight task address spaces. It is unknown if all three, or even any, of the algorithms can be included in the current software architecture without seriously affecting throughput for even one in/one out channel operations.

8.5 SIMULTANEOUS PLAYBACK AND RECORDING FROM MORE THAN ONE INPUT CHANNEL

The previous discussion has been concerned with an increase in the number of transforms applied to any single input/output channel. The final topic of discussion is the possible expansion of the system to playback one channel while simultaneously recording from more than one channel. For this discussion, it is assumed that each recording channel would perform only that subset of functions currently implemented.

Two hardware requirements need to be satisfied before any record channels can be added. First, a second LPA is needed, so that (with appropriate changes in code) the input LPA could be run in dedicated mode to acquire input samples at 40 kHz. Second, approximately 32K words of additional memory is needed for each additional recording channel, since the 128 KW currently available at RADC was nearly filled with the tasks to process only one input and one output channel.

If these hardware requirements are met, it would then be possible to attempt recording from more than one input channel. But evaluation of system throughput at system I&T indicated that at most one channel could be added before throughput failure begins to occur. At that point, the input LPA buffers would have to be increased in size to maintain a constant input LPA QIO overhead. But since there is little task address space left within the Record tasks, it may not be possible to increase the LPA input buffer size. It thus appears that the Record task address space will quickly limit the feasibility of simultaneous multiple input channels.

8.6 REVIEW OF EXTENSIBILITY

The previous five sections have discussed the probability of success in implementing each of the additional features required by the original SOW. Though the design phase and software architecture of the delivered ASMS included consideration for these software extensions, experience has shown that: 1) it may not be possible to have acceptable throughput if all functions are provided for a one in/one out channel operation; 2) it is very unlikely that the current subset of functions can work in more than a two in/one out channel operation. Thus, it is almost certainly the case that throughput failure will occur well before all functions are implemented on all channels. This prediction of early throughput failure is consonant with the original analyses provided by HRB-Singer. In fact, that throughput analysis appears, in retrospect, optimistic; for even with fairly highly optimized Fortran code, only a two in/one out channel operation appears attainable.

While it is clear that extreme difficulties would be encountered in attempting to implement all required functions on the PDP 11/70, it is unclear what a viable alternative architecture would be. Certainly the original microprocessor design should be considered; but it should be noted that its use of the LPA's for A/D and D/A conversions implies two extra bus transfers/ sample. Perhaps more elegant design would place an A/D converter on the front end of each of four microprocessors dedicated to input conversion, compression, and editing; and a D/A converter on the back end of a single microprocessor dedicated to output silence reconstruction, decompression, and conversion. Independent of the final architecture selected, however, it seems clear that some processing must be offloaded onto microprocessors.

9.0 SUMMARY

The Audio Signals Management System evolved through three phases of design. The ASMS implemented at RADC met all requirements of the approved design, as modified by the hardware limitations during System I&T. It is clear that extreme difficulty would be encountered in expanding the existing software to implement all functions described in the second design. At this time, it is unclear what architecture would satisfy those further requirements.

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